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OBUST VOCODER RATE CONTROL IN A PACKET NETWORK

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ROBUST VOCODER RATE CONTROL IN A PACKET NETWORK

BACKGROUND OF THE INVENTION

The range of services offered by wireless communication networks continues its evolution from essentially voice-only service to a rich combination of data services in addition to voice service. One consequence of this evolution is that large portions of the wireless communication network are increasingly designed with an emphasis on supporting the newer, higher bandwidth data services. Wireless Internet access in support of web browsing and streaming media services are examples of these higher bandwidth data services.

In keeping with the nature of these newer data services, the wireless communication network is increasingly packet oriented. For example, a wireless communication network may be, at its core, an assemblage of various network entities interconnected through a packet-based network. While this arrangement suits the packet data flowing between the communication network and the Internet or other packet data networks, it sometimes poses special challenges for legacy services, such as voice.

For example, to reduce the amount of data carried internally by the communication network, voice encoding and decoding (vocoding) functions may be transferred from the radio access network (RAN) to a gateway device, such as a media gateway, that connects the RAN to the Public Switched Telephone Network (PSTN). Voice data received from the PSTN at the gateway device for mobile stations supported by the RAN is compressed and formatted into voice frames, which are then transferred to the RAN in packetized form via some type of packet network interconnecting the RAN and the gateway device.

Locating vocoding functions remote from the RAN imposes special challenges when the RAN needs to assert vocoding control in support of signaling operations. For example, one approach to transferring signaling information to a mobile station is referred to as dim-and-burst, and involves applying greater compression to the voice data so that a voice frame has "room" for one or more signaling bits. Thus, a number of rate-constrained voice frames may be used to transmit a signaling message from the RAN to the mobile station, but only if the RAN has some mechanism for generating or at least requesting the generation of such rate-constrained frames.

Controlling the vocoder in support of dim-and-burst signaling is straightforward when the RAN performs vocoding, but is more complicated when a remote network entity performs the vocoding. When vocoding is remote from the RAN, the network, must have a reliable mechanism for remote vocoder control.

SUMMARY OF THE INVENTION

The present invention comprises systems and methods for controlling vocoding functions that are implemented remote from the radio access network (RAN). For example, a media gateway may interface the RAN with the PSTN and provide vocoding functions for voice data incoming from the PSTN. When the RAN needs to send signaling messages to a mobile station it is supporting, it sends a control message to the media gateway specifying both a constraint rate and a frame count that the media gateway uses to temporarily constrain the rate of one or more voice frames. This allows the RAN to insert signaling information into these rate constrained frames using dim-and-burst signaling techniques.

When the RAN needs to send a signaling message to a mobile station engaged in a voice call, it generates a control message with the appropriate rate constraint and frame count values. Upon sending this control message to the media gateway, the RAN

starts a timer. If the media gateway successfully receives the control message, it will begin applying the requested rate constraint, or possibly a greater constraint, to a defined number of subsequent voice frames encoded for the mobile station by the media gateway.

The RAN inserts the signaling information for the mobile station into these rate-constrained frames using dim-and-burst signaling techniques. If all of the signaling information is sent before expiration of the timer, the timer is stopped and readied for subsequent use. However, if the RAN does not receive a number of rate-constrained voice frames from the media gateway sufficient to transmit all of the required signaling information from the RAN to the mobile station before expiration of the timer, it switches to blank-and-burst signaling. With blank-and-burst signaling, voice data that would otherwise be carried in voice frames transmitted from the RAN to the mobile station is replaced with signaling information.

At the media gateway, any number of control messages may be accumulated and prioritized. A first-in-first-out (FIFO) buffer might be used to accumulated message, for example. With this arrangement, the media gateway reads control messages from its buffer and applies them to the required number of subsequent voice frames sent from the media gateway to the RAN for the involved mobile station or stations. The media gateway may constrain voice frame encoding at the rate specified in the control message, or may form one or more voice frames with a greater constraint applied.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a diagram of an exemplary wireless communication network.

Fig. 2 is a diagram of exemplary vocoding details relevant to the network of Fig. 1.

Figs. 3A and 3B are diagrams of exemplary flow logic for remote vocoder control from the perspective of the radio access network in Fig. 1.

Figs. 4A and 4B are diagrams of exemplary flow logic for remote vocoder control from the perspective of the media gateway in Fig. 1.

Figs. 5A-C are diagrams of exemplary control message formats used in remote vocoder control.

Fig. 6 is a diagram of signaling messages and corresponding vocoding and transmission timing controls.

Fig. 7 is a diagram of control message buffering and corresponding vocoding controls as might be used in the media gateway of Fig. 1.

DETAILED DESCRIPTION OF THE INVENTION

Fig. 1 is an exemplary wireless communication network generally referred to by the numeral 10. The network 10 provides communication between a mobile station 12 and the Public Switched Telephone Network (PSTN) 14 (or other external communication network). The network 10 comprises a radio access network (RAN) 16 and a core network (CN) 18. The RAN 16 interfaces a plurality of mobile stations 12 with the CN 18 and comprises a radio base station (RBS) 20 and a base station controller (BSC) 22. Various entities within the CN 18 provide call setup and processing support for the RAN 16, including a mobile switching center (MSC) server 24, and a media gateway 26, which are all interconnected together and to the RAN 16 by a packet core network (PCN) 28. It should be understood that the network 10 might in practice include various other entities not illustrated, and might include pluralities of one or more entities, illustrated or not.

Fig. 2 illustrates some of the above entities in more detail and provides a convenient basis for discussing operation of the network 10 in the context of voice calls

involving the mobile station 12. The mobile station 12 receives audio input from a user or other audio source, which is converted into digital format and encoded for transmission to the RAN 16. If the network 10 operates in accordance with TIA/EIA/IS-95 or IS-2000 standards, input voice is digitally encoded into twenty millisecond voice frames. The mobile station 12 transmits these voice frames to the RAN 16, which passes them along to the media gateway 26 through the PCN 28 for decoding and transfer to the PSTN 14.

The mobile station 12 includes a vocoder (voice encoder/decoder) 30 that performs the required encoding for voice frames sent to the RAN 16 and decoding for voice frames received from the RAN 16. Voice frames transmitted from the RAN 16 to the mobile station 12 may originate from a number of sources, including the media gateway 26. For example, the mobile station 12 might be engaged in a call with a user of the PSTN 14, in which case the media gateway 26 receives incoming voice data from the PSTN 14, which it then encodes into voice frames, which are transferred to the RAN 16 through the PCN 28, and then transmitted to the mobile station 12.

In support of this role, the media gateway 26 includes a vocoder 32, which might comprise one or more processors 34, and buffer memory 36. It should be appreciated that one or more digital signal processors (DSPs) may be adapted to provide vocoding functions in support of call processing for a plurality of mobile stations 12 engaged in calls with the PSTN 14.

Voice frames sent from the media gateway 26 to the RAN 16 are received by the BSC 22, which passes them along to the appropriate RBS 20 for radio transmission to the mobile station 12. The BSC 22 must also send signaling information (control information) from time to time to the mobile station 12. The nature of this signaling information and the frequency with which it must be sent will depend upon the air interface standard employed by the network 10, as those skilled in the art will readily

appreciate. As was earlier mentioned, the IS-95 and IS-2000 Code Division Multiple Access (CDMA) air interface standards are exemplary references.

Two approaches to sending signaling messages from the RAN 16 to the mobile station 12 are of interest in the context of the present invention. Better appreciating the differences between these approaches requires more detail regarding voice frame encoding. For a given voice call, one of a number of defined rate sets might be adopted. The term "rate set" refers to the maximum voice rate associated with encoding voice data for that call. Examples of typical encoding rates in the IS-95/2000 context are roughly 14.4 kbps, 9.6 kbps, and 4.8 kbps. The encoding rate refers to the effective number of digital bits per second that are used to represent the voice data.

A higher bit rate corresponds to less encoding and to higher voice quality. Thus, for a given call, 14.4 kbps might be set as the full-rate encoding value. The full-rate may vary for different users, and the encoding rate might shift back and forth between full-rate (14.4), half-rate, quarter-rate, and so on, as needed during the call. The need for constraining the encoding rate to something less than full rate, which represent the best voice quality for the given rate set, might arise because of the need to send signaling messages to the mobile station 12, for example.

This point returns the discussion to the signaling formats of interest with regard to the present invention. With dim-and-burst signaling, the encoding rate is constrained to something less than full-rate encoding. This action means that fewer bits of information are used to carry voice information within the rate-constrained voice frames than would be used in full-rate voice frames. Reducing the number of bits given over to voice information leaves "extra" bits available in each voice frame, which bits are used to convey signaling information to the mobile station 12.

Therefore, if the RAN 16 has a signaling message that it needs to send to the mobile station 12, it might simply send a portion of that message in each of a number of

rate-constrained voice frames transmitted to the mobile station 12. While constraining the encoding rate does degrade voice quality somewhat, dim-and-burst signaling usually results in less degradation than arises with the second signaling technique, which is referred to as "blank-and-burst" signaling.

With blank-and-burst signaling, the signaling information replaces all of the voice information that would otherwise be carried within one or more voice frames. Consequently, an entire voice frame is "lost" from the perspective of the receiving vocoder where that frame is blanked by signaling information. While inferior to dim-and-burst signaling from a voice quality perspective, it is sometimes necessary to use blank-and-burst signaling. For example, blank-and-burst signaling might be necessary where transmission of the desired signaling message from the RAN 16 to the mobile station 12 cannot be delayed.

When vocoding functions for the voice frames sent from the RAN 16 to the mobile station 12 reside within the BSC 22, then controlling encoding rates in support of dim-and-burst signaling is straightforward. However, it makes more sense minimize data overhead by transporting compressed voice (encoded voice) through the PCN 28. Accomplishing this data reduction however requires that voice data incoming from the PSTN 14 or other outside network be encoded at the media gateway 26, rather than at the BSC 22. This architectural arrangement requires that the BSC 22 have some mechanism by which it controls vocoding operations in the media gateway 26.

This remote vocoder control is further complicated by the fact that packet networks may occasionally drop data packets. Thus, vocoder control information sent by the BSC 22 is subject to loss within the PCN 28. Such packet data loss might be particularly problematic if the BSC sends a first data packet to initiate rate-constrained encoding at the media gateway 26, and then sends a second packet to end the constrained condition. Loss of the second packet would result in an undesirable

continuation of the rate-constraint condition in the media gateway 26, even if it eventually returns to full-rate encoding by virtue of some time-out mechanism.

Using packet acknowledgement schemes, such as where the commands to enter and exit constrained mode would require some type of ACK or NACK signaling to insure delivery of vocoder control packets might provide the sort of control certainty that is desirable. However, this approach adds too much signaling overhead thereby defeating the original purpose of locating vocoder functions in the media gateway 26. The present invention provides robust vocoder control without need for ACK or NACK signaling, and includes fallback procedures for insuring that signaling messages are sent via blank-and-burst techniques if attempts to send the information via dim-and-burst signaling fail.

Figs. 3A and 3B illustrate flow logic for an exemplary approach to robust control of the vocoder functions in the media gateway 26. The logic flows generally represents the program functions for remote vocoder control in support of call processing associated with mobile station 12. It should be understood that this or similar logic may be used to support vocoder control for a plurality of mobile stations 12. That is, the BSC 22 might independently control encoding rates for many mobile stations 12 to provide those mobile stations with required signaling information.

The BSC 22 may include processors(s) 40, supporting timers 42 and counters 44, and associated memory 46 that support the following functionality. It should be understood that as used herein, the terms “timer” and “counter” encompass hardware and software implementations, and thus should not be construed as necessarily representing some fixed logic circuit or circuits. Indeed, timers 42 and counters 44 may be implemented as software functions by the processor 40, may be actual circuits, such as memory and/or logic circuits, or may be some combination thereof. Further, it should be understood that memory 46 may provide working space for timer and counter functions.

In an exemplary embodiment, processors 40 logically comprise at least processors 40A and 40B, which cooperate in remote vocoder control and mobile station signaling operations. For example, processor 40B might generate air interface signaling messages, or receive them from another entity within the BSC 22, while processor 40A might provide the corresponding control messages to the media gateway 26. Of course, this implementation represents just one of many possible processing embodiments. It should be understood that processors 40A and 40B, or like sets of processors 40A-1..N, and 40B-1..N, may represent logical instantiations of desired processing functions within one or more processing devices or systems generally designated as processors 40.

In an exemplary arrangement, processor 40A supports the flow logic of Fig. 3A, where processing begins (step 100) with the BSC 22 determining whether it has any signaling information for mobile station 12 (step 102). If not, processing continues monitoring for the need to send such information (i.e., step 102 repeats). If there is a signaling message to be sent, the BSC 22 determines a rate constraint value and a corresponding number of frames sufficient to convey the message to the mobile station 12 using dim-and-burst signaling (step 104).

The BSC 22 then sends or transfers a control message comprising the rate constraint and frame count values to the media gateway 26 (step 106). Figs. 5A-5C illustrate exemplary control message structures. Control messages are preferably passed from the RAN 16 to the media gateway 26 in voice frames sent from the BSC 22 to the media gateway 26. Thus, the control message may be structured as a set of binary values. Fig. 5A illustrates one approach, where "V" is a one-bit value that alerts the media gateway 26 to the presence of "CR" and "CL" values within a voice frame. Here, CR and CL represent constraint rate and constraint length values, which tell the media gateway 26 what rate constraint to use and the number of frames to which that constraint applies.

Figs. 5B and 5C illustrates exemplary binary encoding for the rate constraint and constraint length values. Here, both CR and CL are defined as two-bit binary values, and thus may be used to represent any one of four rate constraint values and any one of four frame count values. It should be understood that a greater or lesser number of bits might be used, depending upon the number of unique CR and CL values desired.

By including the constraint rate and constraint length (frame count) values within the same control message, the media gateway receives both the rate-constraint and a corresponding frame count value specifying the number of voice frames to which it should apply the rate constraint. Configuring the control message thusly guarantees that if the media gateway 26 receives it, the media gateway 26 knows both what rate constraint to apply and for how long to apply it. The media gateway 26 will not operate in the rate-constrained condition any longer than necessary to accomplish dim-and-burst signaling at the BSC 22.

However, because the media gateway 26 might not receive the control message at all, or might not comply with it for one or more reasons, the BSC 22 starts a timer 42 in conjunction with sending the control message, buffers the signaling message, and may clear an associated counter (step 108). The timer 42 is configured with an expiration period matched to the time requirements of the signaling message, or may be configured to a default timing value based on other signaling timing requirements. Further, the setting of the timer 42, or the subsequent monitoring for rate-constrained voice frames from the media gateway, may be adjusted to accommodate any network latency or transport delay. That is, there may be a known minimum delay between requesting rate-constrained frames and their subsequent receipt. In any case, once timer 42 is started, initial processing associated with the current signaling message at processor 40A returns (step 102).

Fig. 3B illustrates exemplary flow logic for processor 40B in conjunction with the activities of processor 40A above. In this exemplary embodiment, processor 40B controls transmission of a signaling message based on whether a sufficient number of rate-constrained voice frames are received in timely fashion from the media gateway 26. Processing begins (step 110) with the BSC 22 determining whether there is a signaling message to be sent (step 112). If not, the BSC 22 continues monitoring for signaling messages. Here, monitoring might entail processor 40B receiving a signaling message directly or indirectly from processor 40A, or checking whether a signaling message is otherwise buffered and available for processing. Signaling messages may be processed directly or processed from a buffer in memory 46 in the BSC 22 based on a time priority, a message priority, or a combination of priorities.

If one or more signaling messages are buffered or otherwise available (step 112), the BSC 22 monitors for receipt of rate-constrained voice frames from the media gateway 26 (step 114). If a rate-constrained frame is received before expiration of the timer 42, the BSC 22 sends at least some of the signaling message to the mobile station 12 in that rate-constrained frame using dim-and-burst signaling (116). For each rate-constrained frame received before expiration of the timer 42, the BSC 22 increments one of the counters 44 (step 118), thereby tracking how many rate-constrained frames are received. If the number of rate-constrained frames received matches the frame count value calculated by the BSC 22 (step 120), the signaling message will have been successfully sent. In this case, the BSC 22 stops the timer 42 (step 124), which prevents its expiration, optionally clears the counter 44, and processing returns to monitoring for additional signaling messages (step 112).

If no rate-constrained voice frames are received (step 114), the BSC 22 checks for expiration of the timer 42 (step 128). If the timer 42 has expired, the BSC 22 uses blank-and-burst signaling to transmit the signaling message to the mobile station 12

(step 130). This action prevents delaying transmission of signaling messages from the RAN 16 to the mobile station 12. That is, the timer 42 serves as a fail-safe mechanism in that it allows a suitable period of time in which dim-and-burst signaling may be used, but overrides that signaling scheme with blank-and-burst signaling at the end of that period.

Because the media gateway 26 might apply a greater rate constraint than that specified in the control message (i.e., apply 1/4 rate encoding when 1/2 rate encoding was requested), the signaling message might be sent in a lesser number of frames than the frame count value. Thus, the BSC 22 tracks transmission of the signaling information comprising the signaling message, and checks to see whether the full signaling message has been sent, even if the frame count check is not satisfied (step 122). If message transmission is completed, the timer 42 is stopped to prevent its expiration (step 124) and processing returns to monitoring for signaling messages (step 112). If the message transmission is not completed, and timer 42 has not expired (step 128), processing returns to checking for receipt of rate-constrained frames (step 114).

If a sufficient number of rate-constrained voice frames to support sending the entire signaling message before expiration of the timer 42, the BSC 22 uses blank-and-burst signaling (step 130) to send any remaining portion of the signaling message. Thus, the BSC 22 adopts an approach to signaling where dim-and-burst techniques are preferably used in transmitting signaling messages to mobile stations 12, but where timing safeguards insure timely delivery of those signaling messages using blank-and-burst signaling if necessary.

Each signaling message sent from the RAN 16 to the mobile station 12 generally has its own timing requirements. Because of this, the BSC 22 may maintain separate sets of timers 42 and counters 44 for each signaling message. Indeed, the BSC 22 may

maintain separate logical processes supporting remote vocoder control for signaling operations associated with a plurality of mobile stations 12.

Fig. 6 illustrates an exemplary control configuration at the BSC 22, comprising at least processors 40A-1 and 40B-1, and at least one associated data set 50-1. Some portions of data set 50-1 may be implemented in memory 46. In an exemplary approach, data sets 50 are realized in one or more buffers formed in memory 46. In this manner, processors 40A and 40B can cooperatively write to and read data from these buffers.

In one exemplary embodiment, processors 40A-1 and 40B-1 cooperate to generate and process data within data set 50-1 for one or more mobile stations 12. In an alternate exemplary embodiment, remote vocoder control is implemented on a per-mobile station basis. In this implementation, processors 40A-1 and 40B-1 use data set 50-1 to provide remote vocoder control for a first mobile station 12, while processors 40A/B-2..N and corresponding data sets 50-2..N are used to provide remote vocoder control for additional mobile stations 12.

Regardless of the particular implementation, the BSC preferably maintains separate timers 42 and counters 44 for each signaling message associated with each mobile station 12. In this manner, the BSC 22 ensures that each signaling message is sent according to its priority relative to other signaling messages, or according to some other desired priority scheme, such signaling message age.

Figs. 4A and 4B illustrate complementary logic flows at the media gateway 26 that support remote vocoder control. As with the RAN 16, the media gateway 26 might use this or similar logic to support remote vocoder control for a plurality of mobile stations 12 supported by the RAN 16.

In Fig. 4A, processing begins (step 150), with the media gateway 26 determining whether or not there are any incoming control messages from the RAN 16 (step 152). If

one or more control messages are received, the media gateway 26 buffers the received messages (step 154). If no messages are received or in conjunction with buffering any received messages, the media gateway 26 continues monitoring for incoming control messages (step 156).

The media gateway 26 may receive a plurality of control messages in association with one or more mobile stations 12. Control messages may be buffered and serviced in the order received. Fig. 7 illustrates an exemplary approach, where processor 34A-1 and processor 34B-1 provide control message processing and vocoder control functions. In some embodiments, processors 34A-1 and 34B-1 provide vocoding control for voice frames associated with a plurality of mobile stations 12. Thus, processor 34A-1 queues control messages from different mobile stations 12 in the buffer 36-1, which comprises some or all of memory 36 in the media gateway 26. Processor 34B-1 then retrieves messages from the buffer 36-1, and rate-constrains voice frames for the corresponding mobile stations 12 in accordance with the control messages.

In other embodiments, processors and buffers are logically grouped, such that each group serves a given mobile station 12. In this embodiment, processors 34A/B-1 and buffer 36-1 support vocoding control operations responsive to control messages received in association with control signaling at the BSC 22 for a first mobile station 12. Similarly, processors 34A/B-2..N and associated buffers 36-2..N provide vocoding support for an additional number of mobile stations 12. Thus, the illustrated logic may execute in parallel for a plurality of mobile stations 12.

Of course, this arrangement may represent logical instantiations of processing and buffering functions rather than physically separate processing functions. That is, the set of processors 34A/B-1..N and corresponding buffers 36-1..N may be a logical arrangement rather than a physical arrangement within the media gateway 26. Further,

note that buffers 36-1..N might adopt first-in-first-out (FIFO) buffering, or might adopt some other queuing scheme.

Fig. 4B illustrates operations after receiving a control message, or when a control message is otherwise buffered and available for processing. These operations may repeat until all buffered control messages are processed, and may, as noted, execute in parallel for a plurality of mobile stations. Processing begins (step 160) by determining whether a control message is available for processing (step 162). In an exemplary embodiment, for a given mobile station 12, processor 34A-1 receives and buffers control messages in accordance with the logic of Fig. 4A discussed above, while processor 34A-1B retrieves and processes buffered control messages to provide rate-constrained voice frames.

If no control message is available for processing (step 162), the media gateway 26 continues encoding at the desired rate, which generally implies full-rate encoding (step 164). If a control message is available for processing (step 162), the media gateway 26 changes the encoding rate of the vocoder 32 with respect to the voice frames corresponding to the mobile station 12 with which the control message is associated (step 166). In other words, the media gateway 26 begins rate constraining voice frames intended for the mobile station 12 in accordance with the control message. However, as noted earlier, the media gateway 26 may use a greater rate constraint than was requested by the RAN 16.

The media gateway 26 then transfers one or more rate-constrained voice frames to the RAN 16 (step 168). It will continue sending rate-constrained voice frames until the requested number (i.e., the frame count) of rate-constrained frames is sent, or an equivalent number of more greatly constrained voice frames. For example, if the RAN 16 requested four frames at a 1/2 encoding rate, the media gateway 26 might send four 1/2 rate frames, or might send a fewer number of 1/4 rate frames instead. In either

case, the RAN 16 is provided with enough rate-constrained frames to support its desired dim-and-burst signaling.

In any case, the media gateway 26 tracks the number of rate-constrained frames it sends and determines whether a sufficient number have been sent (step 170). Once a sufficient number of rate-constrained voice frames is sent, the media gateway 26 switches or returns the vocoder 32 to the earlier desired encoding rate (e.g., full rate), or some other desired encoding rate (step 172), and processing returns to monitoring or checking for control messages (step 162). In either case, the media gateway 26 preferably does not use the rate-constrained encoding value any longer than is necessary to support the RAN's dim-and-burst signaling.

While the operating logic described above represents an exemplary approach to robustly controlling remote vocoding functions within the network 10, it should be understood that the present invention permits significant variation. For example, the control message format may be varied, as can the timer/counter techniques for insuring that the RAN 16 timely sends signaling messages to mobile stations 12, whether by dim-and-burst or by blank-and-burst-signaling. Thus, the present invention is not limited by the foregoing description rather it is limited only by the scope of the following claims, and the reasonable equivalents thereof.